Link Adaptation for Streaming Video in EGPRS Mobile Networks

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Abstract— **The poor performance of video compression algorithms in time-varying mobile channels means that the transmission of compressed video over mobile networks is a challenging task. This paper presents a performance enhancement technique for streaming video that makes use of link adaptation. Link adaptation is not generally considered suitable for multi-user streaming video, because it usually requires interaction between the link-layer protocol and the encoder to perform source rate adaptation. The technique presented here requires no such interaction with the encoder, significantly simplifying the link adaptation system. The implementation complexity is very low, enabling a single server to handle large number of users. The benefits of the overall system are demonstrated for MPEG-4 coded video streaming over simulated EGPRS channel conditions. The results reveal that when offered with a similar traffic load, the proposed scheme always provides noticeable improvement of video quality over non-adaptive schemes.**

Keywords-link adaptation, video streaming, EGPRS networks.

I. INTRODUCTION

Several challenges must be addressed to achieve efficient video delivery over cellular networks. Perhaps the most significant problem is the effect of time-varying, high-error, throughput-limited channel conditions experienced by mobile users. To keep performance at an acceptable level, a communication system is traditionally designed for an average or worst-case scenario. However, with worst-case design, the system is severely under-utilized when the channel is in good state. For average case design, poor CIR leads to high variance in video quality and periods of unacceptably poor video output. Such schemes do not provide acceptable quality for all users [1]. An alternative method for limiting the effects of time-varying channel conditions is to employ power control techniques [2]. However, these techniques have a tendency to reduce the overall system capacity due to an increase in interference power to other users.

An alternative to the above is link adaptation. This involves the modification of modulation and coding levels in response to the instantaneous channel and interference conditions. Usually, for best-effort data services, link adaptation algorithms are designed to maximize the overall network throughput [3]. The design of link adaptation algorithms for multimedia services should optimize the delivery of acceptable video quality, facilitating good quality video services with minimum demand on network resources.

Appropriate design of link adaptation algorithms has been shown to result in improved perceptual quality for one-to-one conversational services compared to non-adaptive schemes [4]. In this paper, a method of applying a link adaptation technique to streaming (one-to-many) services is proposed. To achieve link adaptation, the algorithm should ideally be able to switch between a number of source coding rates without performing custom encoding for each user. One way to do this would be to use scalable coding [5]. However, scalable coding tends to be much less compression efficient than a single stream, and is therefore not very efficient in low bit rate channels. Therefore, to improve the efficiency of the link adaptation, the scheme proposed in this paper switches between pre-encoded streams stored at the server. No extra user interaction is required and the link adaptation algorithm is transparent to the end user.

The paper is organized as follows. Section II gives a brief description of the enabling technologies. The proposed link adaptation algorithm is presented in Section III. In Section IV, the simulation set up is described. The algorithm performance is investigated and the performance comparisons are presented in Section V. Finally, Section VI contains the conclusions.

II. EGPRS NETWORKS AND THE MPEG-4 VIDEO CODEC

A. EGPRS System Architecture

Fig. 1 shows how the EGPRS packet-switched system architecture is modified for this proposal. The most important service entities for streaming are the content server and the streaming client, which together with the EGPRS core network entities ensure correct media delivery to the user. Link quality and QoS profile storage is the element that must be added to the system to provide link adaptation. It stores the most current measurements provided by the BSC (Base Station Controller), regarding the time-varying radio link. In addition, it stores hand over and requested QoS parameters for each individual user. This recorded information can be used by a link adaptation algorithm.

Adaptation can be performed by switching between the different EGPRS modulation-coding schemes (see Table I).

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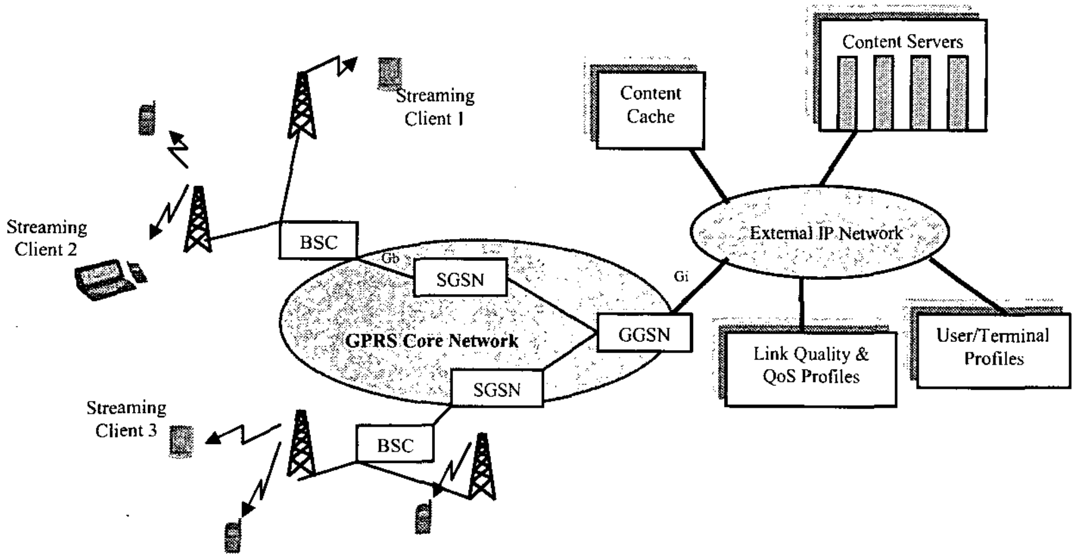


Figure 1. The system architecture: EGPRS packet-switched streaming services.

TABLE I. A SELECTION OF EGPRS MODULATION-CODING SCHEMES .

Scheme	Modulation	Convolutional Code Rate	Data Rate (kbps)
MCS-1	GMSK	0.53	8.8
MCS-2	GMSK	0.66	11.2
MCS-3	GMSK	0.80	14.8
MCS-5	8-PSK	0.37	22.4
MCS-6	8-PSK	0.49	29.6

These schemes employ both Gaussian Minimum Shift Keying (GMSK) and 8-Phase Shift Keying (8-PSK) modulation methods together with convolutional coding schemes with varying rates [6].

B. *MPEG-4 Video Codec*

MPEG-4 is a video-coding standard designed for low bit-rate applications. It provides a number of error resilience tools, which are designed to mitigate errors when operating over error-prone environments such as mobile links and IP-based networks [7]. These tools provide various properties such as error isolation, error resilience, error concealment and data recovery, in order to make the video codec more resilient to channel degradations.

Temporal error propagation may be limited by periodically inserting "Intra" frames. Intra frames do not make use of any information from other frames. These are normally used as refresh frames to clear up errors that may have been propagating through the decoded video sequence. However, Intra frames are much larger than predictively-coded Inter frames, leading to large spikes in throughput, which can result in frame dropping by rate-limitation algorithms. The software developed by the authors makes use of the Adaptive Intra Refresh (AIR) technique described in Annex E.1.5 of the MPEG-4 standard [7]. This forces the insertion of a preset number of non-predictive Intra blocks into each frame. The Intra blocks limit the propagation of errors through successive frames, similarly to Intra frames. But they also remove the need for Intra frame transmission, resulting in much smoother throughput.

III. STREAM SWITCHED LINK ADAPTATION

The proposed Link Adaptation Algorithm (LAA) is designed to improve video quality by varying the source rate and the channel-coding scheme according to measured channel condition, while using a fixed number of EGPRS Time Slots (TS's). This means that although the source rate is allowed to vary, the allocated resources across the radio link do not change. Throughput across the radio interface varies according to the modulation-coding scheme used. This section describes how streams can be switched at the server, and how the link adaptation algorithm operates. Its design is based on the following principles and assumptions:

- Received video quality increases with source rate in an error-free environment.
- As the CIR increases, the received video quality approaches a maximum limit for a given source rate regardless of the channel-coding scheme being used [11].

A. *Stream Switching*

The most obvious way of providing different source rates for each user would be to use scalability. However, this is inefficient in terms of compression. An alternative is to encode "on-the-fly" for each user, which requires significant computation, and does not scale easily. The approach proposed here involves the use of pre-stored streams on the server.

TABLE II. SELECTED VIDEO BIT RATES, R_i, FOR MULTI-SLOT OPERATION IN EGPRS.

Coding Scheme	1 TS (kbps)	2 TS (kbps)	3 TS (kbps)
MCS-1	7.5	15	22.5
MCS-2	9.6	19.2	28.8
MCS-5	19	38	57
MCS-6	25.2	50.4	75.6

The server buffers frames from each stream, and transmits the frame corresponding to the rate specified by the link adaptation algorithm.

Use of the AIR technique (described in Section II) is critical to the success of this switching method. When switching is performed between streams, there is a mismatch between the encoder and decoder in terms of the reference frames used to predict future frames. This can potentially lead to drift [8]. However, the AIR technique limits the effects, and prevents it becoming a detectable problem.

B. Link Adaptation Algorithm

$M \times N$ number of buffers are allocated for each video sequence in the Content Server, where N is the number of Time-Slots in a radio frame (8 for EGPRS) and M is the number of modulation-coding schemes (MCS's) supported (9 for EGPRS). Buffers are labeled:

$$B_{n,m} = \text{where } 1 \le n \le N \text{ and } 1 \le m \le M \qquad (1)$$

According to the video bit rate, R_i, and assuming MCS-5 is used, n number of time-slots, $TS_{n,i}$, are selected for user i. As can be seen from Table II, for selected $TS_{n,i}$, there are M different video source rates (corresponding to different MCS schemes), which can be applied. The link adaptation algorithm is designed to adapt each radio link to one of M MCS schemes. Let $CIR_{k,i}$ be the measured channel condition at the k^{th} radio block for user i. Modulation-coding scheme mode m and video source rate $R_{n,m}$ are chosen if:

$$CIR_{k,i} \in [\xi_{th}(m), \xi_{th}(m+1)] \qquad (2)$$

Where $\xi_{th}(m)$ and $\xi_{th}(m+1)$ indicate corresponding channel threshold values. The main steps of the proposed link adaptation algorithm are listed below.

1. Select n number of TS's needed to satisfy the user's source bit rate requirement.
2. Check for the start of the j^{th} video frame. If start go to step 3.
3. Estimate the channel condition for next radio frame from channel condition measurements. Check for channel-code switching conditions. Select the video source rate $R_{n,m}$ and the MCS for the j^{th} video frame according to the estimated channel condition.
4. Switch to buffer $B_{n,m}$ and use frame header information to find the start of the j^{th} video frame.
5. Transmit data for the j^{th} video frame from buffer $B_{n,m}$ with modulation-coding scheme m using n TS's of the current radio frame.
6. If there are more video frames to be transmitted, return to step 2.

C. Measurement of Channel Conditions

A link adaptation algorithm needs an accurate and recent estimation of the quality of the radio channel. The two main radio link measurements, upon which a channel estimation algorithm is to rely on, are the Received Signal Strength (RSS) and BLock Error Rate (BLER) indicator. Since the RSS is a measure of the total received power, including the noise and interference power, channel quality estimation based on RSS is not very accurate. The systems that rely on the BLER measurements give reasonably accurate channel estimation, although the accuracy is limited by the error detection properties of the channel-protection scheme that is used.

A further concern is the round trip delay. If this delay is too high, the estimated channel condition based on feedback measurements is of little use. In addition to the BLER and RSS, our proposed algorithm uses the first order statistics of the RSS profile in the prediction of channel condition.

Channel estimation is performed by first partitioning a range of possible BLER measurements into a finite number of intervals, I, and then mapping the interval onto an actual channel CIR at the receiver. Measured BLER is a non-linear function of channel CIR and also depends on the channel-coding scheme used. This is because the BLER flag is set if the radio block is found to be in error after channel decoding. Therefore, highly protective channel coding schemes such as MCS-1, which uses a GMSK modulation and 1/2 rate convolutional coding, provide lower BLER than other schemes for the same channel condition. This means that the BLER threshold values should be dependant upon the channel-coding scheme used. Measured BLER is averaged over N radio blocks in order to reduce the effect of burst errors on the channel estimation. Thus, the calculated BLER to be used in the channel estimation algorithm is:

$$BL_{cal,j} = \frac{1}{N}\sum_{k=1}^{N} BL_{k,meas,j} \qquad (3)$$

where $BL_{cal,j}$ is the calculated block error rate. $BL_{k,meas,j}$ is the measured block error rate in k^{th} radio block. Subscript j represents the channel-coding scheme used.

The mean, RS_{mean}, the variance, RS_{var}, and the gradient, RS_{grad}, of the Received Signal Strength (RSS) are used in the channel prediction calculation. The measurement window size is set to be equal to the estimated processing delay, Δ_{est}, which is assumed to be a constant for a given application. Using linear prediction, the predicted RSS is

$$RS_{pre} = RS_{grad} \cdot \Delta_{est} + RS_{mean}. \qquad (4)$$

The predicted RSS is also partitioned into I intervals. Let $CIR_{est}(k)$, $BL_{cal,j}(k)$ and $RS_{pre}(k)$ respectively be the estimated channel conditions, the calculated BLER and predicted RSS at k^{th} radio block. If CIR_i is the corresponding

$$CIR_{est}(k) = \begin{cases} CIR_i, & BL_{cal,j}(k) \in [\eta_{th}(i), \eta_{th}(i+1)] \cap RS_{pre}(k) \in [\mu_{th}(i), \mu_{th}(i+1)] \\ CIR_{est}(k-1), RS_{var}(k) < \gamma_{th} & \\ CIR_l, & BL_{cal,j}(k) \in [\eta_{th}(l), \eta_{th}(l+1)], RS_{pre}(k) \in [\mu_{th}(m), \mu_{th}(m+1)] \end{cases} \quad (5)$$

CIR value for the i^{th} $(i \in \{1,2,......,l\})$ interval, $CIR_{est}(k)$ is estimated according to (5) shown above. Where l<m, and $\mu_{th}(i), \eta_{th}(i)$ and γ_{th} indicate corresponding RSS, BLER and variance of RSS threshold values.

D. Feedback Technique

The feedback channel is used to carry measured channel information, RSS and BLER. Measured RSS is quantized with 8-bits uniform quantization, which provides a resolution of 0.5 dB. The measured BLER is represented by either zero or one. The total 9 bits for each quantized sample is transmitted over the mobile link using a selected modulation-coding scheme. The total information rate on the feedback channel is around 450 bits/s. This is very low compared with the forward user data rate of 28 – 60 kbits/s.

IV. SIMULATIONS

The GSM/GERAN mobile radio multi-path propagation model described in [6] is used to represent the propagation channel. In addition to fast fading and multi-path characteristics, shadow fading with a lognormal variance of 7 dB, and a propagation path loss model were considered in the calculation of the CIR ratio of the channel for different frequency reuse patterns and frequency hopping strategies. Standard (ideal) hexagonal cells and perfect sectorisation are assumed (i.e. no electromagnetic wave penetration to other sectors). The parameters used in the implementation are listed in Table III.

The locations of mobile terminals are uniformly distributed within the cell coverage area and their direction of travel is randomly chosen at initialization. A pseudo-random mobility model with semi-directed trajectories is used to simulate the user mobility. The terminal's position is updated according to the de-correlation length and the direction is changed at each position update with probability of 0.2.

TABLE III. PARAMETERS USED IN THE CHANNEL IMPLEMENTATION.

Parameter	Value
Log-normal variance	7 dB
Decorrelation distance	5 m
Radius of hexagonal cell	200 m
Propagation frequency	900 MHz
Vehicular speed	3 km/h
Channel environment	TU3
CIR margin	9 dB
Fading characteristics	Raleigh fading
Cell configuration	4/12
Frequency hopping	Ideal
Direction change probability	0.2
Path loss model	COST 231-Walfish-Ikegami

Applying the mobility model, the path of the mobile is estimated for the duration of the conversation. The CIR due to the path loss and the shadowing process are simulated separately. The total CIR was estimated by superimposing the simulated shadowing process over the calculated CIR due to path loss.

Experimental results have shown that only MCS-1, MCS-2 and MCS-5 are useful for video, as all of the other schemes result in too high error rates at the decoder to yield acceptable quality [4]. In addition, MCS-1 has a very low bit rate, making it impractical to use without allocating four or more time slots. Therefore, switching in this paper is limited to between MCS-2 and MCS-5, and testing id carried out with 3 time slots. The switching threshold is 22.5 dB, which has been determined from a number of simulations [4].

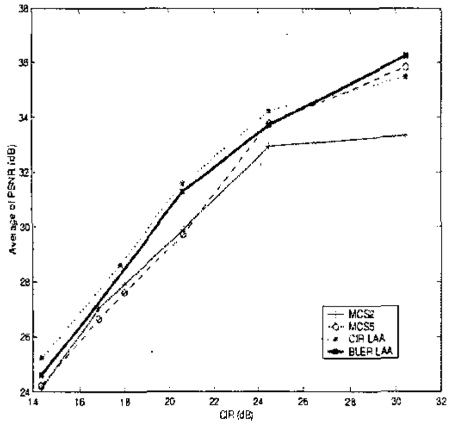
Each video frame is considered as one transport and network layer data payload unit in the EGPRS protocol implementation, as the size of each video frame is below the specified maximum LLC-PDU size (1520 octets) [9] for the bit rates considered.

ITU test sequences "Suzie" and "Foreman" are used as the source signals. QCIF (176 by 144 pixels) sequences are coded at 10 frames/second. To produce longer sequences (25 seconds), these sequences are repeated, with even numbered sequences played in reverse order to guarantee a smooth transition.

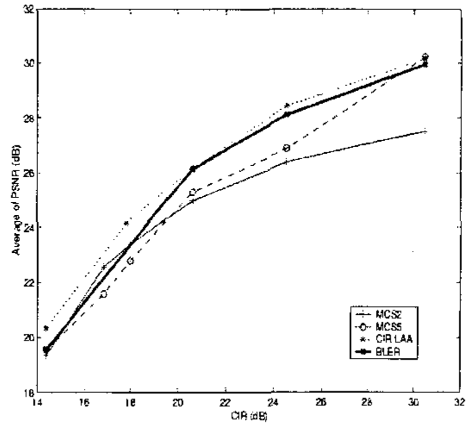
The sequences are MPEG-4 coded using all of the error resilience options and with AIR enabled. Sequences are compressed with different non-varying bit rate outputs using the TM5 [10] rate control algorithm. TM5 varies the quantization to ensure a particular target rate and does not drop frames. To achieve the lower bit rates, it is necessary to reduce the number of non-predictive AIR macro-blocks per frame. This means that while 10 AIR MB's are used for MCS-5, only 5 are used for MCS-2.

V. RESULTS

Fig. 2 shows a comparison of the results for fixed coding and for the proposed link adaptation techniques. Performances are shown in terms of frame Peak Signal to Noise Ratio (PSNR) averaged over 3000 frames. Two different implementation of link adaptation algorithm is considered. The first method uses the actual CIR value of the communication link in stream switching and it is called CIR LAA. Accurate determination of the CIR is difficult in a practical system, so the performance of CIR LAA scheme should be seen as ideal. The second method uses BLER based measurement of channel quality, which is a more practical measurement method for implementation. This is called BLER LAA.



(a) Suzie



(a) Foreman

Figure 2. Simulation results for BLER based LAA.

It should be noted that the CIR value on the x-axis of the graphs is the average CIR of a time varying channel. The results clearly show that both LAA schemes outperform both of the fixed schemes over almost the entire CIR range. The results also show that the performance of BLER LAA is only slightly lower than that of the ideal CIR LAA. LAA schemes improve PSNR by more than 2dB for certain channel conditions. In addition, the minimum required CIR for acceptable quality video reception is improved by at least 2 dB, assuming 30 dB is acceptable for Suzie and 26 dB is acceptable for Foreman. Acceptable quality PSNR is different for both sequences because of the differing levels of compression distortion visible in the two sequences.

Another noticeable element of the results in Fig. 2 is the similar performance of MCS-2 to MCS-5 at low average CIR's. This is due to the lower error recovery properties of the MCS-2 bit stream: each frame contains fewer AIR blocks than in the MCS-5 sequence.

The BLER LAA described here uses channel prediction estimates to minimize the effects of delay on the link adaptation performance. Tests were conducted using a variety of delay values to evaluate the scheme's sensitivity to delay. It is found that the algorithm remains beneficial in the face of typical EGPRS network delays.

VI. CONCLUSIONS

The technique presented in this paper provides a delay robust link adaptation mechanism for sending streaming video over EGPRS mobile networks. The proposed scheme uses the Adaptive Intra Refresh (AIR) technique in MPEG-4 to allow switching between two pre-encoded bit streams, designed for use with two different modulation-coding schemes. This removes the need for encoder-decoder interaction normally associated with link adaptation. By using a prediction method for future channel conditions, the technique is also robust to delay in the feedback channel. Tests were performed using EGPRS channel models, comparing the results of fixed modulation-coding scheme scenarios to two different link adaptation methods. One method uses CIR for determining when to switch, while the other uses BLER. The BLER method is the most practical to implement, and performs only slightly worse than the case using CIR.

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